COURIER 26

June 1976

FRANZ VERTRIEBSGESELLSCHAFT m.b.H.

EMT 250
Electronic Reverberator Unit

EMT at International EXHIBITIONS

10. Tonmeister Convention 75, Nov. 19–22 in Cologne

Equipment on display

A demonstration in the studio

EMT at coming exhibitions	
10–16 September, 1976	Photokina, Cologne
5–9 October, 1976	Ljubljana
1–3 March, 1977	AES Convention, Paris

EMT Courier
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Engineering:

EMT 258

When narrow-band, noisy modulation signals containing significant high frequencies (e.g. single tones from a piano), combined with background noise, are processed through the EMT 258, a burbling sound may possibly be heard: when the filter opens, the desired narrow-band signal, by itself cannot mask the wideband noise which is noticeable as interfering sound. This effect, however, can be avoided if the signal being processed is mixed once again with the unprocessed signal whose level has been reduced by 4 to 6 dB. As a result, the signal-to-noise ratio is indeed improved by only 4 to 8 dB, but this is enough improvement to be noticeable, and the „burbling noise" is completely masked.

EMT 256, 257, 258, 260;

higher input levels.

The EMT 256, 257, 258 and 260 Units are capable of handling input levels up to +6 dBm. However, higher input levels (to +15 dBm) may be acceptable if the input voltage divider is modified. We recommend in this case replacing R 103a and R 104a in the EMT 256, 257, 260 Units (each 4.7 kΩ) with resistors of 15 kΩ each. The EMT 258 on the other hand, should have R 101 and R 102 (each 5.6 kΩ) replaced by resistors of 18 kΩ each.

Who is Who?

Brigitte D. Maecht,

even though she was born in Dresden, has a Hessian accent due to working in Frankfurt for a long time. Before Brigitte came to us, she had worked for American agencies and the U.S. Embassy in Bonn, thereafter in Canada and the U.S.A., a profitable number of years spent in her professional capacity as secretary, including employment at the EMT representative of that time in New York. Since 1961, Miss Maecht has been secretary for Wilhelm Franz. Today, her proven administrative abilities at EMT have led to her full authority for dealing in matters of personnel, auditing and as chief cashier. Additionally, of course, Brigitte's perfect English is useful in many situations.
In private life: a good sort but always knowing exactly what she wants.

A Step Into a New Dimension:

EMT 250

Electronic Reverberator Unit

Fig. 1 View EMT 250

After four years of computer assisted research and development and applying the most modern LSI components plus Schottky high speed TTL logic circuits, EMT is the first to achieve the goal which has for so long been of utmost interest to the audio specialists: high quality reverberation based on true electronics.

The path leading to this attainment was already indicated about 15 years ago when M. Schröder published his article (1). At that time, he was in the position to apply a computer program so that reverberation was added to about 20 seconds of music which was fed into the computer. However, there also was a slight problem: Schröder, at that time, was forced to use a computing speed requiring about 100 times real time. Obviously, executing that procedure in real-time was not possible.

At the 50th AES Convention in London, 1975, B. Blesser and K.O. Bäder reported on a method using a high-speed computer (2) with which it was possible to arrive at a successful real-time conclusion to the Schröder Experiment. With the aid of this computer, and continuous critical listening during the running of the program material, the inadequacies of the Schröder circuit were recognized, improved and finally optimized.

The circuit design resulting from the preceding methods was tested and accepted as final and applied to the EMT 250 Electronic Reverberator Unit.

Because the Unit for producing electronic reverberation must be equipped with a relatively high processing capacity, it is possible to use this capacity for other applications as well. Change-over of the Unit is accomplished solely through changing the internal control program.

Programs:

Reverberation

The program memory is a circuit comprising a configuration of 19 different delay elements each having a different delay time. Some are connected with feedback, the feedback factors being dependent on the switch settings on the control panel. The circuit corresponds

to the following block diagram:

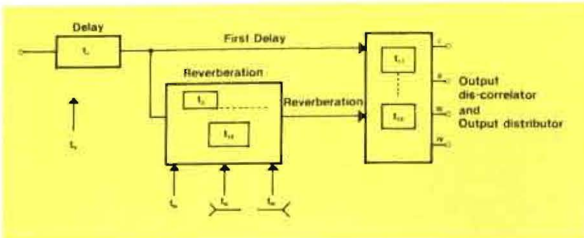


Fig. 2 Block diagram of the electronic reverberator.

This computer optimized circuit, not only produces a very minor coloration of the continuous noise but also a uniform reverberation roll-off.

The outputs are arranged to correspond with the track arrangement for quadrophonic tape recorders (see table).

Output: I	II	III	IV
Channel: left rear	left front	right front	right rear

For stereo, outputs II and III are used; for mono, each output is equally usable.

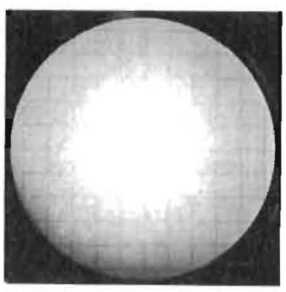


Fig. 3
Oscilloscope photograph of a goniometric display of the reverberation signal; the equally distributed circular-shaped presentation on the scope screen shows an excellent distribution in space.

Delay

This program produces two delay channels. Each of the four outputs can be shifted in steps of 5 ms over the complete range from 0 to 315 ms and without affecting any of the other outputs.

Echo

One variable length delay element is fedback in a way such that the output level is reduced by approx. 1 dB per circulation of the loop. There is a repetitive signal with decreasing intensity: the slap echo program.

The repeating frequency can be varied between approx. 3 Hz (≅ 315 ms) and 200 Hz (≅ 5 ms).

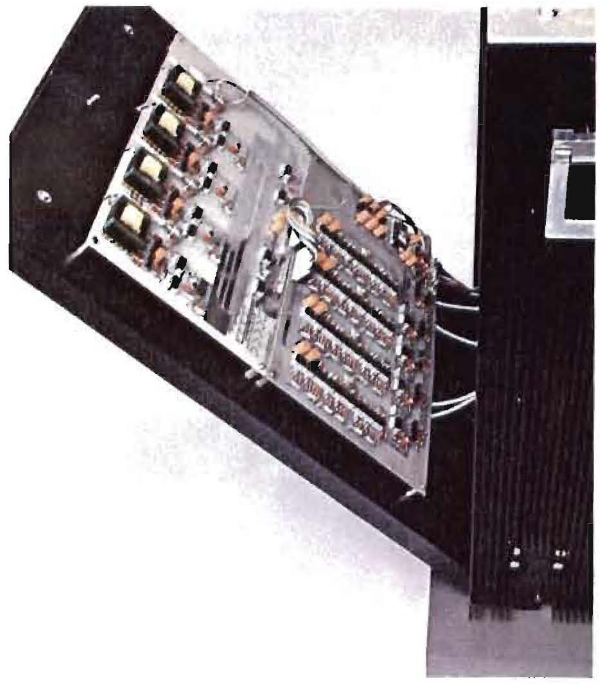


Fig. 4 Audio output board; accessible after tilting out a side section.

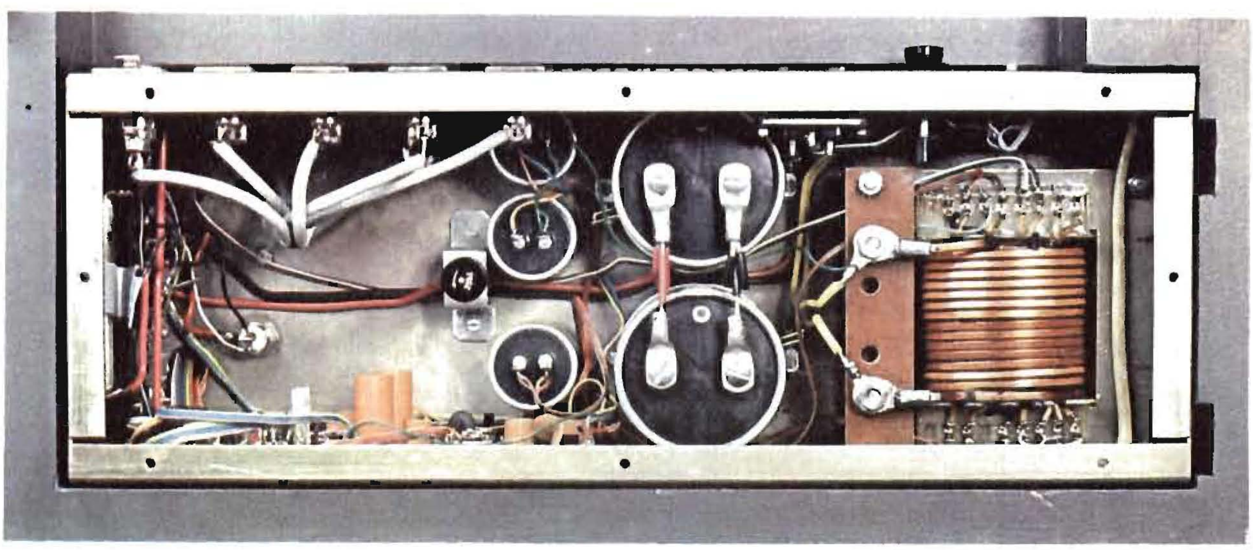


Fig. 5 Power supply; accessible from the bottom of the Unit.

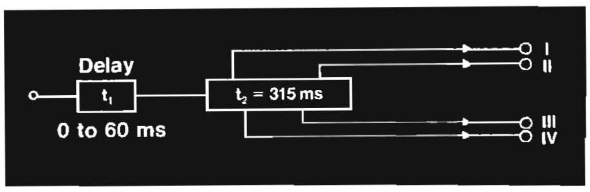


Fig. 6 Block diagram DELAY

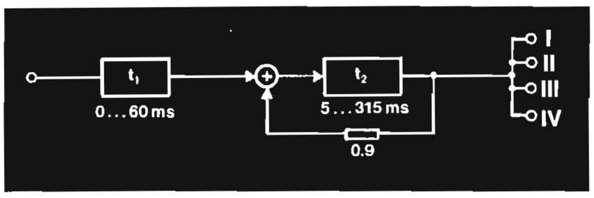


Fig. 7 Block diagram ECHO

„Space"

The program „Space" is a reverberation program of extremely long reverberation time (about 10 s) and with linear distribution of the reverberation time with frequency. Because of atmospheric absorption neither exists in nature, and since the program is intended, amongst others, for science fiction productions, it is designated „Space" (Reverberation in outer Space).

The block diagram of this program basically corresponds to that applied in „Reverberation"; but the number of delay elements and their connections within the true reverberation generator are, in fact, noticeably different. The arrangement of the outputs is according to that of the reverberation program (II and III ≙ stereo).

Chorus Effect

The chorus program results from the consideration that the impression of a large music ensemble is brought about by a certain imprecision, referred to a main microphone. Under the assumption that all musicians in the example illustrated by Fig. 8 are playing absolutely simultaneously, the sound signals originating from each of their positions arrive at the microphone one after the other.

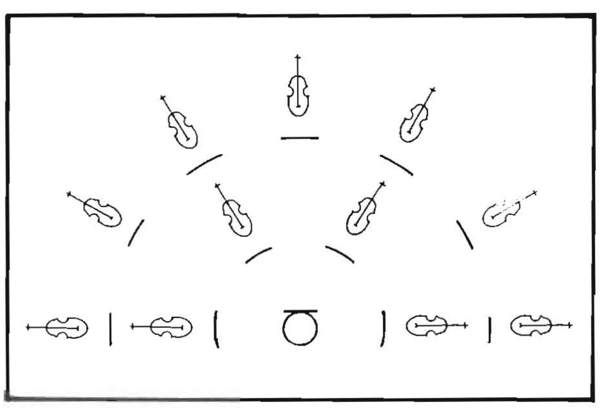


Fig. 8 Arrangement of a string orchestra; there are different arrival times from the individual members to the main microphone.

There are continuous variations in pitch and positions of tones relative to one another. These variations, of course, are very minor, but they are present and are necessary to enable a correct musical impression of a large musical ensemble.

The corresponding block diagram shows four time different delay elements which are being continuously changed through four random signals K_1 to K_4.

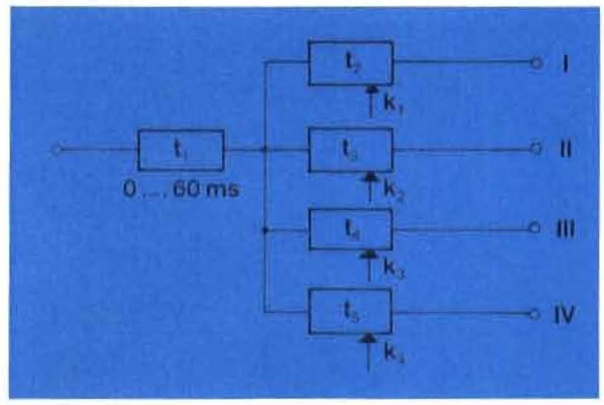


Fig. 9 Block diagram „CHORUS"

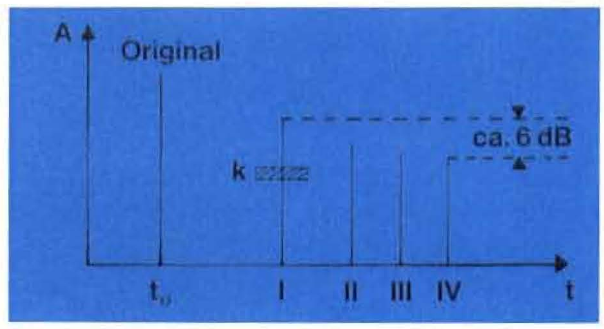


Fig. 10 Time diagram of the individual reflections of the CHORUS program; the delay time varies within the shaded area (k).

Outputs I to IV follow in consecutive order in time and should be mixed together so that the level decreases with increasing delay time.

Stereo Phasing

The phasing effect originates by means of a small shift in time of two signals to one another and by the comb filter curve which is thereby formed.

The block diagram of the circuit used is as follows:

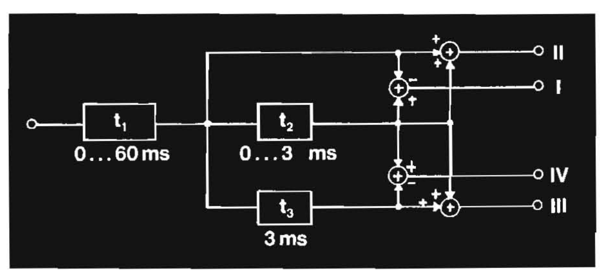


Fig. 11 Block diagram „Stereo Phasing"

Here too, II and III are used as the usual stereo pair. Using the other outputs results in opposing effects.

For mono all outputs can be used. However there are qualifications:
Output I goes to zero when $t_2 = 0$ ms
Output IV goes to zero when $t_2 = 3$ ms

Changing the delay time is done by means of a 16 step switch through a low-pass filter. The speed of the phasing effect, therefore, can be controlled, within a certain framework, through the speed at which the switch is changed.

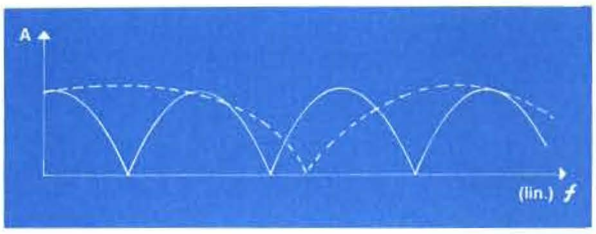


Fig. 12 Comb filter curves of different widths can be produced by means of „Phasing".

Stereo phasing is a new application allowing interesting, shifts of perceptual location to be achieved.

Functions:

The block diagram in Fig. 12 is used for all programs. The input signal is chopped with a pulse repetition frequency of 24 kHz. Every portion resulting from the chopping process is analyzed with respect to its amplitude by a comparator having very high resolution. In twelve discrete steps, it is determined whether the amplitude is located above or below one-half of the comparative value. Thus, twelve YES/NO items of information are produced or $2^{12} \triangleq 4096$ possibilities. These are put out as 12 bit digitally encoded signals.

As soon as the input signal falls 6 dB below full signal drive, the input amplifier gain is raised by 6 dB. This gain change occurs at −6, −12 and −18 dB; it occurs without time delay within the 30 μs sampling period. Within a fraction of this time, the amplifier responds to the new value. However, this change of gain must be compensated at the output of the unit: an inverse gain change is used for compensating. For that purpose, the information relevant to the gain status also in binary coded form written into the shift register, is delayed together with the signal and then is used as correcting variable for the output amplifier.

After the A/D converter, there follows a dynamic MOS shift register of approx. 24 K bits which permits setting a fixed pre-delay of 0, 20, 40 or 60 ms for all programs. Then the digital signal is fed to the actual processor. Owing to the extremely high operating speed of about 6 MHz per process step, it permits approx. 250 processing steps to be executed within the sampling time of 40 μs, during which the digital word is available. Therefore, many calculating operations can be carried out at every step.

The sequences and types of these calculating operations and therewith the selected program are determined by the program memory. It comprises ROM's (Read Only Memory) with a capacity of 16 K bits, addressed in increasing sequence by means of the clock-pulse generator.

Intermediate data are stored in a read/write memory made up of RAM's (Random Access Memory) having a capacity of approx. 125 K bits. If these intermediate data are to be first extracted for use again in later process periods, then by this means the delay chains or delay loops could be set-up and become available for use when needed in the diverse programs.

From the central processor, a D/A converter takes the signals, and in time-sharing mode supplies signal drive to the four different outputs I to IV.

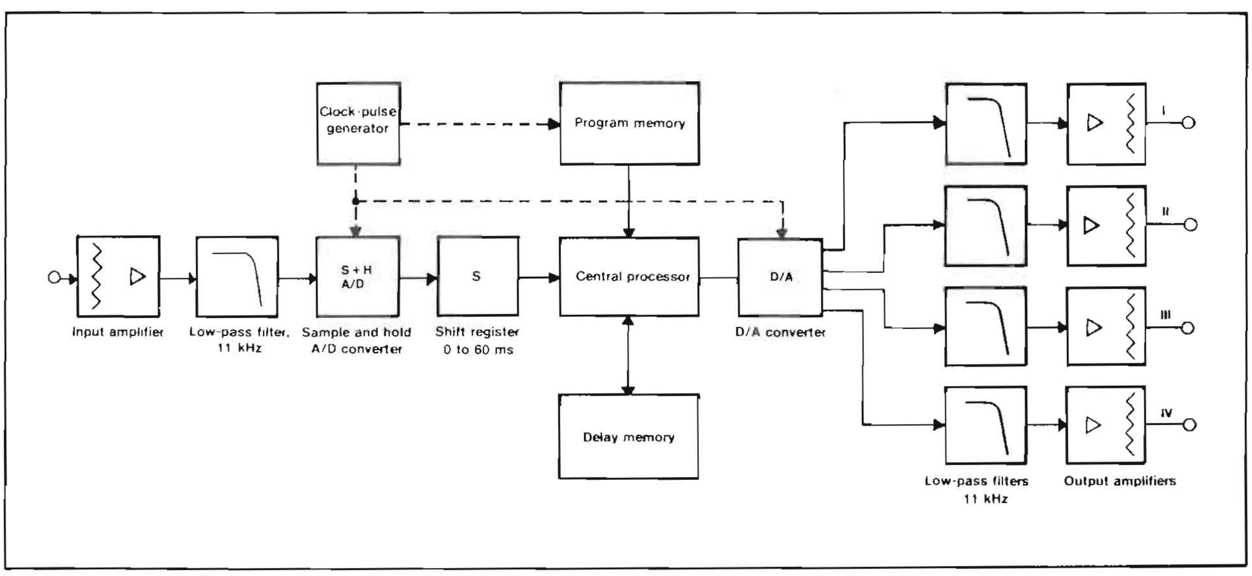


Fig. 13 Block diagram of the complete EMT 250

Physical and Mechanical Setup

The EMT 250 Electronic Reverberator Unit is constructed as a free standing unit. The principles of the mechanical assembly are shown in Fig. 14. The power supply is assembled in a chassis placed at the bottom of the Unit; thereby preventing the AC power line from passing through sensitive sections of the Unit. At the center of the Unit, separated by the internal partition, there is the analog section placed at one side, and the digital section at the other side. Both side panels can be tilted out for servicing purposes; thus assuring excellent access to all electronic components. The four panels of the EMT 250 are made of black anodized extruded aluminium that guarantees perfect heat sink for the heat generated within the power stages.

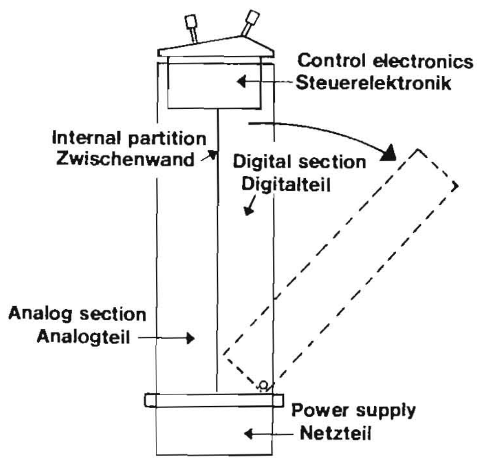


Fig. 14 Sketch of EMT 250 assembly; both side panels tilt out.

At the upper part of the EMT 250, is placed the control electronics with the control hardware on the top panel. This was planned to permit placing the EMT 250 directly next to the audio engineer at the control console. As a result, the control hardware must be arranged so that the engineer can handle the controls without looking at the knobs and switches. For that reason, the four most important switches are of a type of „stick" control (see photo Fig. 15).

The switches serve various purposes according to the set-up program.

With program REVERB:

Main switch, left (16 steps):	reverberation, 0.4 to 4.5 s
Main switch, left (4 steps):	reverberation at bass frequencies, lettered as factors 0.5 to 2 at 300 Hz relative to the position of the main switch
Main switch, center (4 steps):	reverberation at treble frequencies, lettered as factors 0.25 to 1 at 6 kHz relative to the position of the main switch

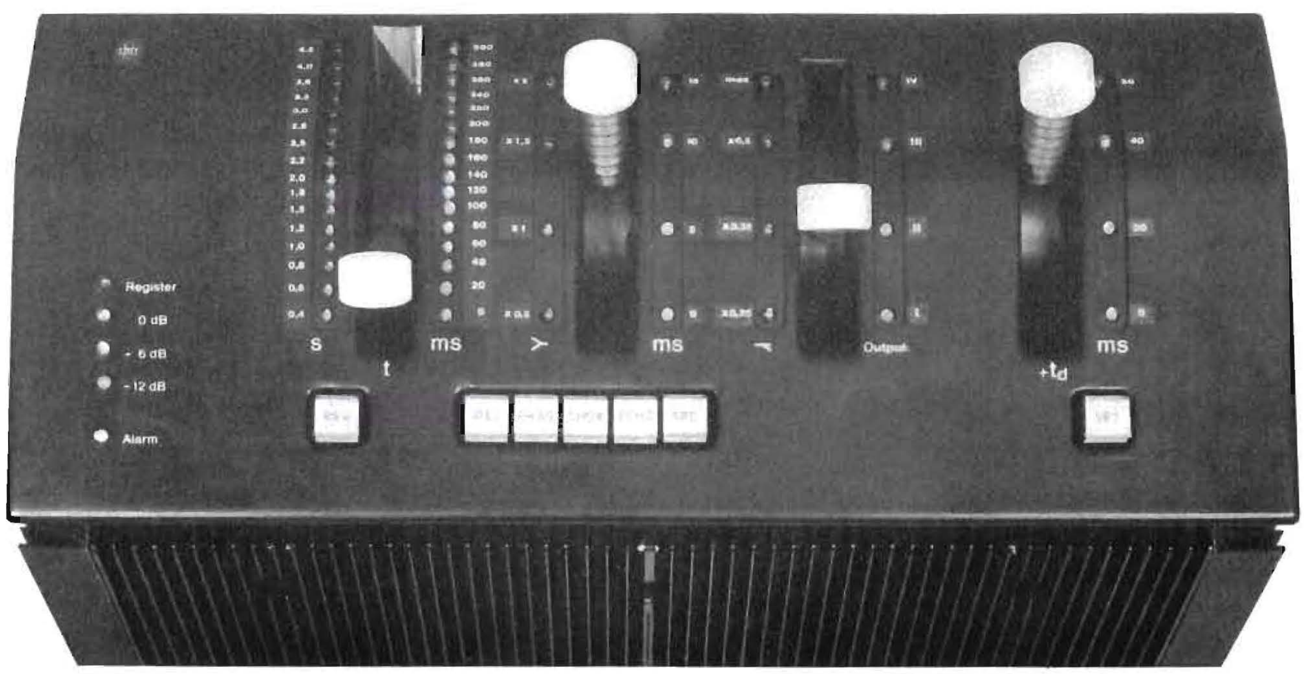
The given values are indicated by **red** LED's to the **left** of the appropriate switch.

With program DELAY:

Main switch, left (16 steps):	delay time, coarse, in steps of 20 ms from 0 to 300 ms
Main switch, left (4 steps):	delay time, fine, in steps of 5 ms from 0 to 15 ms
Main switch, center (4 steps):	programing for the outputs I, II, III and IV.

The given values are indicated by **green** LED's to the **right** of the appropriate switch.

Fig. 15 Operating panel of EMT 250



The program is selected by the push button array below the aforementioned main switches.

The program memory interrupts its cycle if the push button SET has been actuated, and verifies the positions of the program push button and main switches in use at that time. Correspondingly, it then issues new instructions to the processor.

After every change of program or settings, the SET push button must be actuated. The new values are taken-over only after SET has been actuated.

An additional LED lettered ALARM indicates by blinking when the temperature within the Unit has reached a critical value. The Unit, however, is not yet switched off so that the recording can be continued to the end. Only after a further 5°C increase of the internal temperature is the Unit automatically switched off.

Four additional LED's check the signal from the Unit output so that at all times the recording engineer has an overview as to whether the Unit is being fed correctly.

References:

(1) Schröder, M. R.: „Colorless Artificial Reverberation," JAES, July 1961.

(2) Blesser, B. A., Bäder, K. O., and Zaorski, R. „A Real-Time Digital Computer for Simulating Audio Systems". Presented March 5. 1975 at the 50th Convention of the Audio Engineering Society, London, England.

Radio Station WTFM in Fresh Meadows, N.Y., U.S.A.

Three years after the appearance of the fourth generation Flutter Meter, the EMT 424, the new EMT 422 Flutter Meter is available for delivery.

EMT 422 Flutter Meter

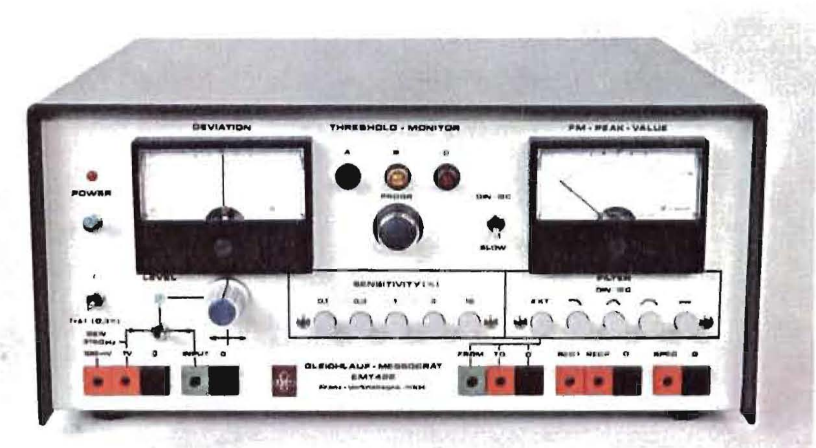


Fig. 1:
Front panel view

We present a compact piece of test equipment which offers, due to its technical details as well as its economical price, special features both for servicing and for production line testing.

Measurements:

Five measurement ranges, from ±0.1% to ±10% full scale deflection, permit measuring the flutter and FIM of mechanisms used in the recording and reproduction of sound. Two panel meters indicate the slow speed deviations (DEVIATION) and the quasi peak values of flutter (FM-PEAK-VALUE).

Flutter evaluation is possible not only by means of the internationally recognized weighting curve having its maximum at 4 Hz, conforming to IEC/DIN/ANSI standards, but also through the additional three filter curves within the EMT 422:

, LINEAR: from 0.2 Hz to 300 Hz (−3 dB)
, Bandpass filter: from 20 Hz to 300 Hz (−3 dB)
, Low-pass filter: roll-off: 6 dB/Octave.

These permit an initial quick analysis of the flutter frequency.

An additional provision has been made for external filters and graphic recorders or oscilloscopes to be connected for aiding in the evalution.

Circuit Details:

The connection data are essentially the same as those for the larger EMT 424 Flutter Analyzer, from which the most important circuits have been taken; e.g. the tone generator, the input stages, and the phase-locked loop (PLL) circuit for evaluating the frequency modulation representing the actual flutter. For the first time, a new feature is incorporated: the so-called **THRESHOLD MONITOR** circuit. This is obtainable as an option (additional printed circuit board):

> By means of programmable plug in modules, any two threshold values may be selected, and each value is associated with an indicator lamp for a certain flutter value coordinated with the reading on the right hand panel meter.
>
> The circuit is arranged so that the first, green lamp A lights immediately when the meter exceeds 5 % of FSD; up to the first selected threshold value of, for instance, 20 % FSD. For higher meter readings, the second, yellow lamp B lights: up to the second threshold value corresponding for example to 70 % FSD. Now the third, red lamp C lights.
>
> By means of two simple resistors in the three pole programming plug, it is possible to select the threshold values for X % or Y % readings, independent of the measuring range being used at the time.

This allows a simplified quality classification for repetitive measurements in maintenance departments and on the production line.

Placing the switch in the SLOW position, dampens the meter movement, which affects both the flutter meter reading and the threshold circuit.

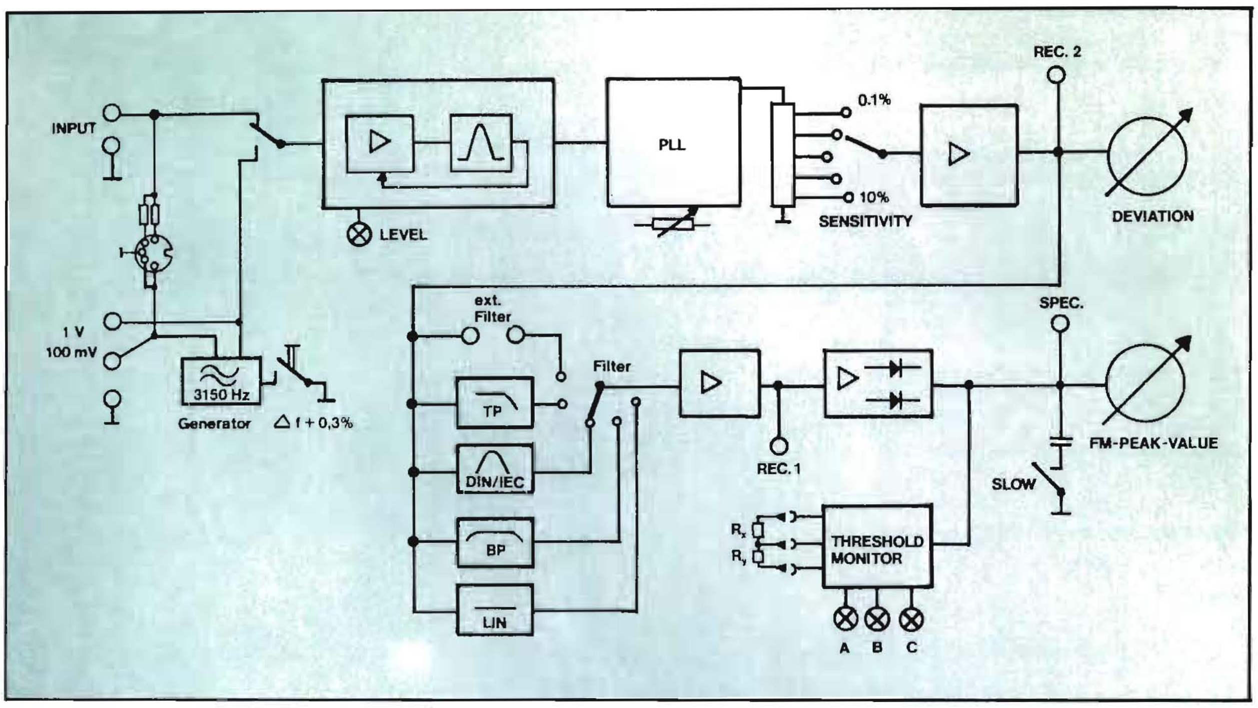


Fig. 2 Block diagram EMT 422

As with the EMT 424 Flutter Analyzer, it is possible to measure directly the frequency intermodulation distortion (FIM) of disk pick-ups according to DIN 45 411 (E).

Block diagram:

As the block diagram shows, the input circuit comprises a compressor-amplifier with a filter in the control loop. Thus, the EMT 422 input level may operate between 40 mV and 3 V, without interference from any amplitude fluctuations which might possibly appear. A LEVEL lamp lights when there is sufficient input level. Following that is an EMT 424 circuit which is already known, the FM demodulation stage using a PLL circuit.

The SENSITIVITY switch and an amplifier stage feeds the DEVIATION meter. It indicates slow speed deviations or slip page in sound recording and reproducing equipment. The same information is available at the output RECORDER 2 for connection to a graphic recorder or an oscilloscope.

The FILTER switch permits selecting 4 internal or an external filter for measuring the peak flutter value.

The output RECORDER 1 is located ahead of the rectifier circuit which has the dynamic characteristics according to IEC/DIN, and after the rectifier is the SPECTRUM output.

Also at this point, the SLOW switch is located which connects a capacitor to damp the meter reading so as to have integrated measurement values.

At this point, the signal is also extracted for the previously described threshold monitor circuit.

The stable 3150 Hz generator delivers two output levels: 1 V and 100 mV. By means of the f + Δf switch, the generator can be detuned by +0.3 % for calibration during recording, or for other test purposes.

A 5-pole DIN connector is mounted on the back panel of the EMT 422. It is connected in parallel with the banana jacks on the front panel and is used for Input/Output connections.

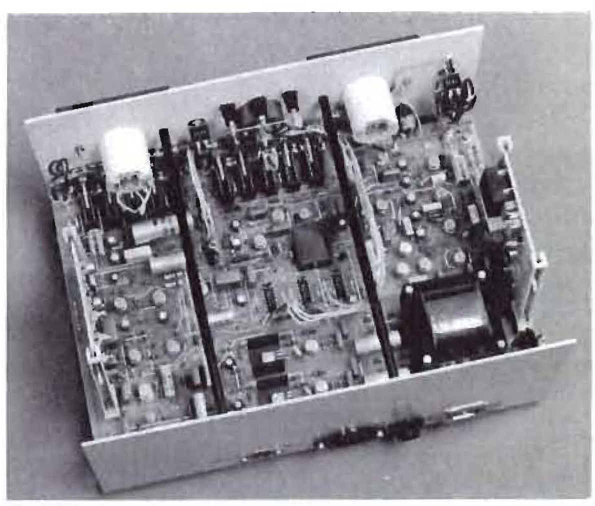


Fig. 3 Interior view of EMT 422

With the exception of the plug-in modules, generator and threshold monitor, the other circuits are located on the basic printed circuit board.

A carrying handle is sunk into the case cover. As an accessory, a plexiglass cover is available for protecting the important operating controls against erroneous operation. For instance, the sensitivity switch might be protected during repetitive production line testing.

Shock absorbing frame for Studio Turntable: to damp mechanical and acoustical vibrations

EMT 930 - 900

Task:

For applications where the EMT 930 Turnable must be operated in rooms subjected to strong mechanical vibrational shocks, we have developed the shock absorbing frames described here. These frames isolate the turntable mechanism against all mechanical and acoustical stimulation. For instance mechanical vibrations brought about by poorly constructed floors, or through console-cabinet designs which cannot meet stability requirements. Additionally, self-resonances in the sheet metal panels must be positively prevented from disturbing the Turntable. Self-resonances could be caused by operating personnel inadvertently introducing shocks and jars or from airborne sound impinging on the panels.

If the signals are monitored through loudspeakers in the same room, then the shock absorbing frames considerably decrease the danger of acoustic feedback.

Construction:

When the shock-absorbing frame is used, the turntable is mounted on a subframe made of a particularly dense material instead of being placed in the usual plywood cutout. Turntable and subframe are then together supported by ellastic mounts. The outer frame is cut out somewhat larger to permit the mechanism and subframe to swing freely. Since there are two different vibration directions; vertical and horizontal, specially designed ellastic elements are provided, with their damping selected to optimize the self-resonance of the suspended system. In the vertical plane, the „Turntable in the Frame" unit rests on four adjustable helical springs. Each of the helical springs is combined with a damping element in a round case along with a guide which permits only vertical movement of the spring and the supporting mount. The damped suspension in the horizontal plane is accomplished by round flexible rubber cords fastened between the supports on the vertically suspended subframe and the fixed main frame. For rough handling during transport, a mechanical stop is provided to block free motion of the subframe.

The described shock absorbing mountings dampen only low frequency vibrations and mechanical shock; the higher frequencies are damped at the same time through rubber elements connecting the subframe and the springs to the supports.

Principle of operation:

If turntables are examined for the effects of external mechanical stimulation, then all disturbances introducing relative movement of pick-up arm and chassis, lead to a disturbing signal at the output. For linear movement, such a system is almost completely insensitive because the pick-up arm bearing is located at the arm's center of gravity. The pick-up arm cannot change position relative to the chassis, even if the position of the chassis is shifted. This, of course, results from the pick-up arm being mounted at its center-of-gravity, and shifting position cannot bring about rotary movement. But another situation is brought about by rotary stimulation. If the chassis is caused to move in a rotary direction, then the pick-up arm, in consequence of its moment of inertia, maintains its position. Such stimulation leads to a self-oscillation in the pick-up arm resonance. Disturbances are observed in the output signal, and if the excursion amplitudes become excessive, the stylus could jump out of the groove.

The main task of a shock absorbing suspension system, therefore, is to prevent rotation of the turntable chassis and to reduce translation motion. Proceeding from there, the essential disturbance components are translation motion, while rotational disturbances appear negligible by comparison.

Through the benefit achieved from low self-resonance, it is possible to assign priority directions to the swinging movement of the chassis; i.e. soft suspension for the two directions in the horizontal plane. Thus, this preferential treatment reduces rotation, and horizontal shocks are absorbed with only minor reaction. It is essential, therefore, that the suspension have no tendency to change of mode. For instance translation should not be converted into rotation. This is achieved through the correct arrangement of the spring fastening points with reference to the center of gravity for the complete system. External forces must always lead to a resultant passing through the center of gravity.

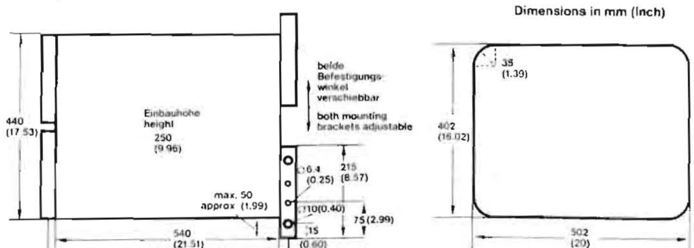
No use is made of suspension elements such as simple helical spring arrangements, which could permit horizontal shocks to be converted to vertical vibrations. For the shock absorbing frames presented herein, horizontal shocks are only permitted to lead to damped horizontal vibrations.

Results:

A test of a sample shock absorbing frame in different rooms and mounted in different cabinets, unequivocally indicates sufficient sound insulation from mechanical vibrations and noises caused by disturbing the cabinet.

For usual disturbances, the affects on the modulation signal lie below the rumble measured with good test records. When two EMT 930 Turntables are directly compared, one with and the other without shock absorbing frames, a 15 dB interference decrease can be measured over the transmission range. This value can vary by a few dB, depending on the frequency position of the disturbing stimulation.

Abmessungen in mm (Inch)
Dimensions in mm (Inch)



vorgeschlagener Ausschnitt der Abdeckmaske
proposed cut-out of top cover

REVERBERATION

from the gold foil with DELAY of the first DIGITAL reflection

The secret of natural sounding artificial reverberation* lies in the 30–60 ms delay of the first reflection and the following reverberation achieved by the two dimensional sound expansion of the plate or gold foil.

EMT 240 + EMT 440

* In natural room acoustics, the first reflection is followed by an infinite number of reflections exponentially decreasing in level and rapidly increasing in density. This is artificially achieved by use of the electronic delay unit EMT 440 with its wear-free digital storage network, together with the infinite reflections of the reverberation foil EMT 240 and its hitherto unattainable resonance density. Variable delay time of 0.8 to 5 seconds results in an incomparable fullness of sound.

ELEKTRONIK, MESS- & TONSTUDIOTECHNIK

EMT-FRANZ VG mbH.

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